AMENDMENTS TO THE SPECIFICATION:

Please amend the paragraph beginning on page 2, line 31, and continuing to page 3, line 3, as follows:

It has become apparent from the above that the conventional automatic speech recognition system 100 depicted in Fig. 1 analyzes the input speech signal in a spectral range up to 4 kHz by sampling the analog input speech signal at 8 kHz. Of course, higher sampling rates may be used as well. For example, personal computers often use a sampling rate of 11 kHz which represents 1/4 of the ~~44,1~~ 44.1 kHz used for the sampling of CDs. It is evident that a higher sampling bandwidth is connected with more spectral information so that the performance of automatic speech recognition systems generally increases if higher sampling rates are employed.

Please amend the paragraph beginning on page 3, line 21, and continuing to page 3, line 26, as follows:

The speech analysis in this network system is based on a MEL filterbank with 23 subbands. The number of 23 MEL subbands is kept constant for all three sampling rates. This means that the subbands are differently distributed over each of the three spectral ranges of 4, ~~5,5~~ 5.5 and 8 kHz (corresponding to the sampling rates of 8, 11 and 16 kHz) to be analyzed.

Please amend the caption on page 4, line 9, as follows:

BRIEF SUMMARY ~~OF THE INVENTION~~

Please amend the paragraph beginning on page 4, line 11, and continuing to page 4, line 18, as follows:

~~According to the invention, a~~A speech analyzing stage of an automatic speech recognition system ~~is proposed for analyzing~~analyzes a speech signal sampled at one of at least two different system sampling rates in the spectral domain~~, t~~. The speech analyzing stage ~~comprising~~ comprises a first spectral analyzer for analyzing the speech signal up to a first frequency and a second spectral analyzer for analyzing the speech signal at least above the first frequency.

Please amend the paragraph beginning on page 4, line 20, and continuing to page 4, line 26, as follows:

A method ~~according to the invention~~ for analyzing in the spectral domain a speech signal sampled at one of at least two different system sampling rates of an automatic speech recognition system comprises a first analysis step for analyzing the speech signal up to a first frequency and a second analysis step for analyzing the speech signal at least above the first frequency.

Please amend the paragraph beginning on page 4, line 28, and continuing to page 4, line 37, as follows:

~~According to the invention, a~~At least two spectral analyzers are provided, each spectral analyzer analyzing the speech signal in the spectral domain. The first spectral analyzer analyzes the speech signal in a lower spectral range having an upper frequency limit which is defined by a first frequency. The first frequency is preferably derived from the lowest system sampling rate. The lowest system sampling rate is the lowest sampling

rate occurring in an automatic speech recognition system in which at least two different sampling rates are utilized.

Please amend the paragraph beginning on page 5, line 13, and continuing to page 5, line 21, as follows:

The first spectral analyzer ~~according to the invention~~ ensures a high compatibility among the components of an automatic speech recognizing system working with several system sampling rates since for all sampling rates a compatible set of acoustic parameters can be obtained. This compatible set of acoustic parameters is generated by the first spectral analyzer which independently from the sampling rate, i.e. even for the lowest sampling rate, always parametrisizes an identical spectral range up to the first frequency.

Please amend the paragraph beginning on page 5, line 35, and continuing to page 6, line 2, as follows:

A further advantage ~~of the invention~~ is the fact that the recognition stage of an automatic speech recognition system can be simplified because the compatible set of acoustic parameters allows to perform pattern matching using one and the same pattern matching unit for acoustic parameters sampled at different sampling rates.

Please amend the paragraph beginning on page 9, line 19, and continuing to page 9, line 28, as follows:

The ~~invention~~ <u>technology</u> also relates to a data signal to be transmitted from the terminal to the network server having a central speech recognition stage, the data signal comprising a first data structure relating to the sampling rate and a second data structure containing at least one codebook index derived from a codebook for a specific

combination of one or more acoustic parameters obtained by analyzing the speech signal up to a first frequency and one or more further acoustic parameters obtained by analyzing the speech signal at least above the first frequency.

Please amend the paragraph beginning on page 10, line 4, and continuing to page 10, line 11, as follows:

The ~~invention~~ technology can be implemented for example as a hardware solution and as a computer program product comprising program code portions for performing the individual steps of the invention when the computer program product is run on an automatic speech recognition system. The computer program product may be stored on a computer readable recording medium like a data carrier attached to or removable from a system component.

Please amend the paragraphs beginning on page 10, line 28, and continuing to page 11, line 12, as follows:

Fig. 3 is a block diagram of an automatic speech recognition system comprising a first embodiment of a speech analyzing stage ~~according to the invention~~;

Fig. 4 is a block diagram of a distributed speech recognition system ~~according to the invention~~;

Fig. 5 is a block diagram of a second embodiment of a speech analyzing stage ~~according to the invention~~ which may be used in the DSR systems of Fig. 4;

Fig. 6a to 6c are block diagramms showing the use of codebooks in speech analyzing stages ~~according to the invention~~; and

Fig. 7 is a schematic diagram of a data signal ~~according to the invention~~ to be used in the DSR system of Fig. 4.

Please amend the paragraph beginning on page 12, line 24, and continuing to page 12, line 33, as follows:

The problems encountered with the automatic speech recognition system depicted in Fig. 2 are overcome by the automatic speech recognition system 10 depicted in Fig. 3 and comprising a first embodiment of a speech analyzing stage 12 ~~according to the invention~~. The automatic speech recognition system 10 further comprises a recognition stage 14 with a single pattern matching unit 16. The pattern matching unit 16 performs pattern matching based on the acoustic parameters received from the speech analyzing stage 12 and based on reference models which are stored in a database not depicted in Fig. 3.

Please amend the paragraph beginning on page 13, line 22, and continuing to page 13, line 33, as follows:

The first spectral analyzer 18a arranged in the first speech analyzing branch 12a of the speech analyzing stage 12 is configured to analyze the speech signal in a spectral range up to f_{lowest}. This upper spectral boundary of the spectral range analyzed by the first spectral analyzer 18a was derived from the lowest system sampling rate 2 x f_{lowest} by

multiplying the lowest system sampling rate 2 x f_{lowest} by ~~0,5~~ 0.5. The upper spectral boundary of the spectral range analyzed by the first spectral analyzer 18a could also be chosen such that it equals less than half the lowest system sampling rate. As an example, if the lowest system sampling rate is 8 kHz, f_{lowest} could equal ~~3,5~~3.5 kHz.

Please amend the paragraph beginning on page 16, line 4, and continuing to page 16, line 8, as follows:

~~In~~ Fig. 4~~,~~ depicts an example DSR system 200 ~~according to the invention is depicted~~. The DSR system 200 comprises a single network server 210 with a central recognition stage 210a. The network server 210 communicates via wired or wireless communication links 212 with three terminals 214, 216, 218, e.g. mobile telephones.

Please amend the paragraph beginning on page 17, line 14, and continuing to page 17, line 20, as follows:

As can be seen from Fig. 5, the second spectral analyzer 18b analyzes the spectral range between 4 and ~~5,5~~ 5.5 kHz and outputs M (M ≥ 1) additional acoustic parameters relating to the speech energy in this spectral range. The third spectral analyzer 18c analyzes the spectral range between ~~5,5~~ 5.5 kHz and 8 kHz and outputs N (N ≥ 1) additional acoustic parameters relating to the speech energy in this spectral range.

Please amend the paragraph beginning on page 18, line 19, and continuing to page 18, line 27, as follows:

The further terminal 216 operated at a sampling rate of 11 kHz could be constructed using a speech analyzation stage 216a with only the first two speech analyzing branches 12a, 12b depicted in Fig. 5 since a speech signal sampled at 11 kHz

does not comprise spectral information above ~~5,5~~5.5 kHz. For the same reason the speech analyzation stage 214a of the terminal 214 operated at a sampling rate of 8 kHz would only require a single speech analyzing branch similar to the speech analyzing branch 12a depicted in Fig. 5.

Please amend the paragraph beginning on page 21, line 12, and continuing to page 21, line 20, as follows:

In Fig. 6b, determination of the codebook index for the acoustic parameters c_1 and c_2 obtained at a sampling rate of 11 kHz within the terminal 216 is illustrated. The codebook 28b deviates from the codebook 28a depicted in Fig. 6a in that the codebook 28b comprises a further column $E_{4-5,5}$ relating to the speech energy contained within the speech signal in a frequency range between 4 kHz and ~~5,5~~ 5.5 kHz. The value of $E_{4-\cancel{5,5}\underline{5.5}}$ is determined by a speech analyzing branch similar to the speech analyzing branch 12b depicted in Fig. 5.

Please amend the paragraph beginning on page 21, line 33, and continuing to page 22, line 3, as follows:

In Fig. 6c, determination of the codebook index for the acoustic parameters c_1, c_2, $E_{4-\cancel{5,5}\underline{5.5}}$ and $E_{\cancel{5,5}\underline{5.5}-8}$ obtained within the terminal 218 at a sampling rate of 16 kHz is illustrated. The codebook 28c depicted in Fig. 6c deviates from the codebook 28b depicted in Fig. 6b in that it comprises a further column for the acoustic parameter $E_{\cancel{5,5}\underline{5.5}-8}$. The codebook index for the quadruplet of acoustic parameters c_1, c_2, $E_{4-\cancel{5,5}\underline{5.5}}$ and $E_{\cancel{5,5}\underline{5.5}-8}$ is determined in a similar manner as described above with reference to Fig. 6b.

Please amend the paragraph beginning on page 23, line 4, and continuing to page 23, line 27, as follows:

In Fig. 7, the overall data structure of a data signal 50 to be transmitted from one of the terminals 214, 216, 218 to the network server 210 is depicted. The data signal 50 comprises a first data structure 52 which contains synchronization information and header information relating to the sampling rate at which the corresponding speech signal has been sampled. A second data structure 54 of the data signal 50 contains a plurality of data substructures 56 to 68. The first data substructure 56 relates to the logarithmic frame energy which was linearly quantized with eight bits. The six remaining data structures relate to the codebook indices for the six pairs of acoustic parameters c_1 to c_{12} (see above table). The second data substructure 58 contains the seven-bit codebook index which was generated as explained above with reference to Figs. 6a to 6c. This means that the codebook index contained in the second data substructure 58 was derived from one of the codebooks 28a, 28b, 28c for a specific combination of the first two acoustic parameters c_1, c_2 and the one or two further acoustic parameters $E_{4\cancel{-5,5}\underline{5.5}}$, $E_{\cancel{5,5}\underline{5.5}-8}$ which were obtained by analyzing the speech signal in upper frequency ranges. The five further data substructures 60 to 68 depicted in Fig. 7 relate to the five further pairs of acoustic parameters depicted in the above table. Altogether, the seven data substructures 56 to 68 contain 44 bits of information.

Please amend the paragraph beginning on page 24, line 23, and continuing to page 26, line 2, as follows:

From the table it can be seen that for HMMs trained at a sampling rate of 16 kHz a high recognition accuracy (word error rates smaller than ~~1,07~~<u>1.07</u> %) can be expected independently of the sampling rate of the speech signal. Moreover, a gain can be seen when moving from the lowest sampling to higher sampling rate and applying HMMs

trained at the respective sampling rate. No further gain can be achieved when moving from 11 kHz to 16 kHz. This is not surprising because there exists only little spectral information in speech signals above ~~5,5~~5.5 kHz. From the table it also becomes clear that independently from the sampling rate of a terminal a high recognition performance can be obtained for all HMMs.